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**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

| Office Action Summary | Application No. | Applicant(s) | |
|---|---|---|---|
| | 10/783,437 | KUBLER ET AL. | |
| | Examiner | Art Unit | |
| | HOANG-CHUONG Q. VU | 2476 | |

*-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --*

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☒ Responsive to communication(s) filed on 21 September 2009.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) 22-40 and 42-69 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 22-40, 42-69 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

## DETAILED ACTION

**Status of Claims**: Claim 41 has been cancelled. Accordingly, claims 22-40, 42-69 are currently pending.

### *Claim Rejections - 35 USC § 102*

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

2. **Claims 22, 27, 28, 34, 37-40, 43, 45-47, 52-55, 57-60, 65-69** are rejected under 35 U.S.C. 102(e) as being anticipated by Baum et al. (5,761,281).

**Regarding claim 22**, Baum et al. disclose a communication system controller comprising: interface circuitry for communicating, with an information transmission device, at least one of information requesting setup of a call and parameters for configuring the information transmission device **(see col. 5 lines 50-60; control signals are exchanged between the network application module (network application module includes all the circuitry shown in Fig. 7A-B) and the modem for configuring the modem)**; wherein the parameters for configuring the information transmission device comprise information related to the conversion of digitized voice information into an analog voice signal, and an analog voice signal into digitized voice information **(see Fig. 1; modem inside network server is configured with**

**configuration parameter (claim 1) to support conversion between digital and analog as shown in Fig. 1) (data as taught by Baum before being placed on the T1 line is in analog from and transmitted on the telephone lines 40, 42, and 44. The calls from the call originators are then converted to digital signals (calls are digitized) and placed into the T1 line (see col. 8 lines 1-7)) (call signaling is for voice service (col. 19 lines 30-31) and the signaling bits (for T1 DS0) translate directly to E and M signaling (signaling for voice service) (col. 19 lines 65-66))**; at least one processor communicatively coupled to the interface circuitry **(see Fig. 7A-B; CPU 633 in the network applicant module 82)**; and operational software executable by the at least one processor, the operational software **(see col. 9 lines 24-32; the network application module 82 functions as an interface and includes software to process control signals to configure the modem)** causing the at least one processor to produce the parameters for configuring the information transmission device based upon the information requesting setup of a call **(see col. 30 lines 47-50; protocol parameters for configuration of the modem are produced by converting the control signals; where the control signals is extracted from the an incoming call; the call setup and the modem configuration are performed using the extracted control signals (see col. 3 lines 1-12))**, the information transmission device thereby communicatively coupling one of a plurality of communication networks to another of the plurality of communication networks **(see Fig. 1 and col. 6 lines 34-36; the configured modem inside server 30 communicatively coupling telephone company network 50 to local area (token ring) network 52)**.

**Regarding claim 27**, Baum et al. further teach wherein the plurality of communication networks comprises a conventional telephone switching network **(see Fig.1; PSTN 50)**.

**Regarding claim 28**, Baum et al. further disclose the conventional telephone switching network communicates using analog signals **(see col. 8 lines 1-2; data transmitted on the telephone lines at 40, 42, and 44 is in analog form)**.

**Regarding claim 34**, Baum et al. further teach wherein the operational software is capable of determining a routing for the requested call **(column 21 lines 49-54; routing decision for the incoming call is based on ANI/DNIS information extracted from the control signals)**.

**Regarding claim 37**, Baum et al. further teach wherein the information requesting setup of a call comprises information related to telephony signals received by the information transmission device **(col. 5 lines 24-35)**.

**Regarding claim 38**, Baum et al. further teach wherein the telephony signals received comprise at least one of dual tone multi-frequency (DTMF) signals, dial tone, a ring signal, on-hook, off hook, and call progress tones **(col. 20 lines 26-34)**.

**Regarding claim 39**, Baum et al. further teach wherein the parameters for configuring the information transmission device comprise information related to telephony signals generated by the information transmission device **(col. 25 lines 3-27; multi-frequency tone is looked up in the memory to retrieve DNIS digit and to find specific parameter information using the DNIS; thus parameter is related to the tone exchanged between network application module and the modem)**.

**Regarding claim 40**, Baum et al. further teach wherein the telephony signals generated by the information transmission device comprise at least one of dual tone multi-frequency (DTMF) signals, dial tone, a busy signal, and a ringing signal **(col. 20 lines 26-34)**.

**Regarding claim 43**, Baum et al. further teach wherein the parameters for configuring the information transmission device comprise information related to at least one of a battery supply, over-voltage protection, ringing current, tone generation, tone detection, two wire to four wire conversion, and test functionality **(see col. 5 lines 11-20; parameter related to error correction technique (test function is performed before error correction). Also see col. 16 lines 48-52)**

**Regarding claim 45**, Baum et al. further teach wherein the interface circuitry is capable of communicating digitized voice information with the information transmission device **(see col. 5 lines 54-55; digital data exchanged between the modem and the network application module; also see col. 8 lines 5-9; calls (voice) from the call originator are digitized (digital data) and fed into the network server 30)**.

**Regarding claim 46**, Baum et al. further teach wherein the communication system controller and the information transmission device are located within the same housing **(see Fig. 2; network application module and the configured modem are located with in the chassis of the network server 30)**.

**Regarding claim 47**, Baum et al. disclose a communication system controller comprising: interface circuitry capable of providing configuration information to a system supporting the communicative coupling of one of a plurality of communication networks

to another of the plurality of communication networks based upon the configuration information **(see Fig. 1 and col. 6 lines 34-36; the configured modem inside server 30 communicatively coupling telephone company network 50 to local area (token ring) network 52) (col. 30 lines 47-50; where the modem is configured using the protocol parameters produced from converting the control signals)**; wherein the configuration information comprise information related to the conversion of digitized voice information into an analog voice signal, and an analog voice signal into digitized voice information **(see Fig. 1; modem inside network server is configured with configuration parameter (claim 1) to support conversion between digital and analog as shown in Fig. 1) (data as taught by Baum before being placed on the T1 line is in analog from and transmitted on the telephone lines 40, 42, and 44. The calls from the call originators are then converted to digital signals (calls are digitized) and placed into the T1 line (see col. 8 lines 1-7)) (call signaling is for voice service (col. 19 lines 30-31) and the signaling bits (for T1 DS0) translate directly to E and M signaling (signaling for voice service) (col. 19 lines 65-66))**; storage capable of containing operational software and call routing information **(see col. 25 lines 8-9; lookup is performed to retrieve DNIS digit in the memory; where the DNIS digit is used to route a call (col. 27 lines 4-5))**; and at least one processor communicatively coupled to the interface circuitry **(see Fig. 7A-B; CPU 633 in the network applicant module 82) (see col. 9 lines 24-32; the network application module 82 functions as an interface and includes software and memory to process control signals to configure the modem)**, the at least one processor

capable of accessing the operational software and call routing information, the operational software functioning at least to cause the at least one processor to produce the configuration information based upon call setup information and the call routing information **(see col. 30 lines 47-50; protocol parameters for configuration of the modem are produced by converting the control signals; where the control signals is extracted from the an incoming call; the call setup and the modem configuration are performed using the extracted control signals (see col. 3 lines 1-12); and the control signals can be correlated to call routing parameters (col. 4 lines 60-63))**.

**Regarding claim 52**, Baum et al. further teach wherein the plurality of communication networks comprises a conventional telephone switching network **(see Fig.1; PSTN 50)**.

**Regarding claim 53**, Baum et al. further disclose the conventional telephone switching network communicates using analog signals **(see col. 8 lines 1-2; data transmitted on the telephone lines at 40, 42, and 44 is in analog form)**.

**Regarding claim 54**, Baum et al. further teach wherein the call setup information is received via one of the plurality of communication networks **(see col. 21 lines 27-31; call set-up is received from the telephone company network).**

**Regarding claim 55**, Baum et al. further teach a network interface adapted to communicate using a wired network **(Fig. 1-2; server 30 includes interface to communicate with the telephone switching network 50. also see col. 19 lines 33-37)**.

**Regarding claim 57**, Baum et al. further teach wherein the call setup information is received via the wired network **(col. 19 lines 33-37 and col. 21 lines 27-31)**.

**Regarding claim 58**, Baum et al. further teach wherein the call setup information comprises a destination address **(see col. 33 lines 55-57; where the incoming call is associated with a destination telephone number called by the call originator)**.

**Regarding claim 59**, Baum et al. further teach wherein the call routing information comprises at least one association of a destination address and a call route **(see col. 33 lines 55-57 and col. 4 lines 60-63; the control signals can be correlated to call routing parameters)**.

**Regarding claim 60**, Baum et al. disclose a machine-readable storage having stored thereon a computer program having a plurality of code sections for implementing a communication system controller, the code sections executable by a machine for causing the machine to perform the operations comprising: storing routing information received from a user at a first location **(see col. 33 lines 29-32; storing communication protocol parameter associated with control signal identified with an incoming communication)**; accepting a call setup request from the user via one of a plurality of communication networks **(see col. 33 lines 33-36; where the server interfacing between a network and the incoming call from the call originator)**, the call setup request comprising a destination address corresponding to a second location **(see col. 33 lines 55-57; where the incoming call is associated with a destination telephone number called by the call originator)**; determining routing information based upon at least one of the call setup request and the stored routing information for

the first user **(see col. 24 line 65 – col. 25 line 9; lookup is performed to retrieve DNIS digit in the memory when detecting a multi-frequency tone start signal; where the DNIS digit is used to route a call (col. 27 lines 4-5))**; generating configuration information using at least one of the call setup request and the routing information **(see col. 30 lines 47-50; protocol parameters for configuration of the modem are produced by converting the control signals; where the control signals is extracted from the an incoming call; the call setup and the modem configuration are performed using the extracted control signals (see col. 3 lines 1-12); and the control signals can be correlated to call routing parameters (col. 4 lines 60-63))**; wherein the configuration information comprise information related to the conversion of digitized voice information into an analog voice signal, and an analog voice signal into digitized voice information **(see Fig. 1; modem inside network server is configured with configuration parameter (claim 1) to support conversion between digital and analog as shown in Fig. 1) (data as taught by Baum before being placed on the T1 line is in analog from and transmitted on the telephone lines 40, 42, and 44. The calls from the call originators are then converted to digital signals (calls are digitized) and placed into the T1 line (see col. 8 lines 1-7)) (call signaling is for voice service (col. 19 lines 30-31) and the signaling bits (for T1 DS0) translate directly to E and M signaling (signaling for voice service) (col. 19 lines 65-66))**; and providing the configuration information to a device capable of communicatively coupling the user via one of a plurality of communication networks to the second location via another of the plurality of communication networks in order to

establish the requested call **(see Fig. 1 and col. 6 lines 34-36; the configured modem inside server 30 communicatively coupling telephone company network 50 to local area (token ring) network 52)**.

**Regarding claim 65**, Baum et al. further teach wherein the plurality of communication networks comprises a conventional telephone switching network **(see Fig.1; PSTN 50)**.

**Regarding claim 66**, Baum et al. further disclose the conventional telephone switching network communicates using analog signals **(see col. 8 lines 1-2; data transmitted on the telephone lines at 40, 42, and 44 is in analog form)**.

**Regarding claim 67**, Baum et al. further teach wherein the determining comprises: determining whether routing information corresponding to the destination address is available using the stored routing information and the destination address **(col. 21 lines 48-51; determining to route calls placed to a particular phone number associated with computers C3-C5)**; prompting the user for routing information, if routing information corresponding to the destination address is not available **(col. 21 lines 54-60; computer C1 provides call setup information)**; and refraining from prompting the user, if routing information corresponding to the destination address is available **(col. 21 lines 58-63; modem converts the signals when computer C5 is available)**.

**Regarding claims 68 and 69**, Baum et al. further teach sending to the second location a call setup request and receiving from the second location acceptance of a call setup request **(see col. 31 lines 23-28)**.

### ***Claim Rejections - 35 USC § 103***

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. The factual inquiries set forth in *Graham* **v.** *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

5. **Claims 23-25, 29-33, 44, 48-51, 56, 61-63** are rejected under 35 U.S.C. 103(a) as being unpatentable over Baum et al. (5,761,281) in view of Henley et al. (5,526,353).

**Regarding claims 23-25, 48, 50-51, and 61-63**, Baum et al. disclose all the subject matter of the claimed invention as recited in claims 23, 48, and 61 respectively without explicitly teach wherein the plurality of communication networks comprises a packet network; wherein the packet network communicates using an Internet protocol (IP) which comprises transmission control protocol (TCP)/Internet protocol (IP). However, it must be noted that Baum et al. disclose the token ring network 52 in which the network can receive digital signal thru the server 30 as shown in Fig. 1. Henley et al. from the same or similar field of endeavor disclose a system and method for communication of audio data over a packet-based network. The teaching recite

Transmission Control Protocol/Internet Protocol (TCP/IP) is one of the supported network and transport protocols **(column 4, lines 6-7)**. Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to implement the packet-based network taught by Henley et al. in the local area network 52 in the teaching of Baum et al. and to include TCP/IP as a transport protocol in the call delivery system as taught by Henley et al. One is motivated as such to employ the teaching of Henley et al. as the domain of digital computer networks continue to expand to carry data representing a digitized audio signal such as voice in the digital network (see Henley et al. col. 4 lines 15-24).

**Regarding claims 29-30, and 56**, Baum et al. disclose all the subject matter of the claimed invention as recited in claims 22 and 55 respectively above without explicitly teach a packet network interface for communicating using a packet protocol wherein the packet protocol is compliant with an Ethernet protocol. However, Henley et al. from the same or similar field of endeavor teach using a system and method for communication of audio data over a packet-based network. Henley et al. recite a preferred embodiment directed to Ethernet environment where each node in the computer network is designated by a specific address **(column 6, lines 15-21)**. Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Baum et al. to make the protocol compliant with an Ethernet protocol. One is motivated as such to enable each packet assembly circuit the ability to determine the routing of the audio data through the network with a packet-based transmission protocol **(column 6, lines 27-31)**.

**Regarding claims 31 and 49**, Baum et al. further disclose packets communicated comprise digitized voice information **(see col. 8 lines 5-9; calls from the call originator are digitized and fed into the network server 30)**. Henley et al. also disclose this limitation by providing digital service of audio data from the Ethernet physical layer **(column 9, lines 36-40)**.

**Regarding claims 32 and 33**, Baum et al. further teach wherein the packets communicated via the packet network interface comprise non-voice data; wherein at least a portion of the non-voice data is unrelated to the communication of digitized voice information **(see col. 5 lines 36-38; packets exchanged related to credit card transaction (non-voice data) an unrelated to digitized voice information)**.

**Regarding claim 44**, Baum et al. disclose all the subject matter of the claimed invention as recited in claim 22 above without explicitly teach reducing the quantity of digitized voice information communicated via the information transmission device, by changing the packetization of digitized voice information when voice activity on one of the plurality of communication networks falls below a predetermined level. However, it must be noted that Baum et al. teach configuring the modem with protocol parameter which provides modulation scheme, data compression, and transmission rate to properly handle framing, filtering, and forwarding to serve the call (col. 5 lines 8-19). Henley et al from the same or similar field of endeavor teach a system and method for communication of audio data over a packet-based network. It is disclosed the system further comprises a decimation circuit for deleting audio data from a designated location of the buffer to shorten the portions of the stream of audio data in the buffer. The circuit

addresses the problem when data are read from the buffer slower than they are written to the buffer **(column 5, lines 65-67 and column 6, lines 1-5)**. Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Baum et al. to reduce the quantity of digitized voice information communicated via the information transmission device by changing the packetization of digitized voice when voice activity on one of the plurality of communication networks falls below a predetermined level. One is motivated as such to ensure the buffer stays close to its predetermined length for efficient realignment of the audio data in the buffer (column 6, lines 11-14).

6. **Claims 26, 64** are rejected under 35 U.S.C. 103(a) as being unpatentable over Baum et al. (5,761,281) in view of Henley et al. (5,526,353), and further in view of Lev et al. (5,729,544).

**Regarding claims 26 and 64**, Baum et al. and Henley disclose all the subject matter of the claimed invention as recited in claims 23 and 61 above respectively without explicitly teach wherein the packet network comprises a wireless network. However, Lev et al. from the same or similar field of endeavor teach wherein the packet network comprises a wireless network **(see col. 2 lines 62-63)**. Thus, it would have been obvious to one of ordinary skill in the art at the time the invention was made to employ the wireless network as taught by Lev et al. in the teaching of Baum et al. and Henley et al. to provide addition coverage to wireless customers. The motivation or suggestion would have been to extend the use of applications to remote locations not serviced by LAN/WANs (see col. 1 lines 19-20).

7. **Claim 35** is rejected under 35 U.S.C. 103(a) as being unpatentable over Baum et al. (5,761,281) in view of Barak (5,764,741).

**Regarding claim 35**, Baum et al. disclose all the subject matter of the claimed invention as recited in claim 34 above without explicitly teach wherein the routing is determined based upon a cost of use of a communication network. However, Barak from the same or similar field of endeavor teaches wherein the routing is determined based upon a cost of use of a communication network **(see Abstract lines 2-8; determining routing based on the cost information in the routing database)**. Thus, it would have been obvious to one of ordinary skill in the art at the time the invention was made to use routing cost in a routing database taught by Barak to determine which providers or networks to execute the call. One of ordinary skill in the art would have motivated to do so to select a least cost route for a call.

8. **Claim 36** is rejected under 35 U.S.C. 103(a) as being unpatentable over Baum et al. (5,761,281) in view of Fleischer, III et al. (5,592,541).

**Regarding claim 36**, Baum et al. discloses all the subject matter of the claimed invention as recited in claim 34 above without explicitly teach wherein the routing is based upon predefined call routing information. However, Fleischer, III et al. from the same or similar field of endeavor teach wherein the routing is based upon predefined call routing information **(see col. 1 lines 9-12; routing call based on predetermined routing options)**. Thus, it would have been obvious to one of ordinary skill in the art at the time the invention was made to employ the predetermined routing options as taught by Fleischer III, et al. in the teaching of Baum et al. to forward calls based on the

individual needs. One of ordinary skill would have motivated to do so to allow subscribers to define and tailor their telecommunication services (see Fleischer III et al. col. 1 lines 6-8).

9. **Claim 42** is rejected under 35 U.S.C. 103(a) as being unpatentable over Baum et al. (5,761,281) in view of Sharman (5,774,854).

**Regarding claim 42**, Baum et al. discloses all the subject matter of the claimed invention as recited in claim 22 above without explicitly teach the parameters for configuring the information transmission device comprise information related to the buffering of digitized voice information for a predefined period of time to minimize gaps in an analog voice signal. However, Sharman from the same or similar field of endeavor teaches a text to speech system operating in real using an acoustic processor and a linguistic processor. Due to the computational time the linguistic processor requires to process data, future requests from the acoustic processor cannot be made. Thus gaps in the speech output often occur when the acoustic processor requests data from the linguistic processor. Sharman proposes a solution to overcome the gaps in data by adjusting the buffer for minimal of output data so that future requests can be supplied in a timely manner **(column 7, lines 39-48)**. Hence the propagation delay caused by the linguistic processor is a factor affecting the adjustment in the buffer for desired optimal output. Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Baum et al. to have the parameters configuring information related to the buffering of digitized voice information for a predefined period of time in order to minimize gaps in the analog voice signal as

taught by Sharman. One is motivated as such to accurately halt the system based on the output in the event that an interruption occurs (abstract, column 2, lines 34-39).

***Response to Arguments/Remarks***

10. Applicant's remarks/arguments filed September 21, 2009 have been fully considered but they are not persuasive.

11. On pages 15-19 of the Applicant's remarks regarding claim 22, Applicant submits that the cited art does not teach each and every element of Applicants' claim 22. Applicants' claim 22 recites information requesting setup of a call and parameters for configuring the information transmission device and the parameters are produced for configuring the information transmission device based upon the information requesting setup of a call. Applicants understand the Office to be asserting that "control signals" of Baum teach Applicants' claimed "parameters for configuring the information transmission device". In view of the rejection, Applicants have noted that the Examiner does not identify the teaching of Baum that corresponds to Applicants' claimed "information requesting setup of a call.

The Examiner respectfully disagrees because the Examiner has equated the control signals as equivalent to the claimed "information requesting setup of a call". For further clarification, when a call originator initiates a call, the call originator provides call set-up information including telephone number and the control signals (col. 21 lines 48-60). The control signals can be in the form of multifrequency tones identified with the telephone number called by the call originator (col. 5 lines 31-35). Thus, when the call originator initiates a call and provides set up information, the call originator requests

routing decision for the incoming call based on the information extracted from the control signals (col. 21 lines 52-55). Therefore the Examiner has shown that the information provided by the call originator is the information for setting up a call and the call originator requests a call by initiating the call with control signals provided for setting up the call.

The Examiner has further shown “parameters for configuring the information transmission device are produced based upon the information requesting setup of a call”. Evidently, col. 30 lines 47-50 of Baum suggests that protocol parameters for configuration of the modem are produced by decoding multifrequency tones and converting the control signals.

12. On pages 19-21 of the Applicant’s remarks, Applicant has amended claim 22 to further include aspects of claim 41. Applicant disagrees with the Examiner’s assertion that Baum discloses parameters for support conversion between digital and analog. Applicant submits that those of ordinary skill in the art at the invention would recognize that T1 line is a digital communication path. The Examiner agrees that T1 is a digital communication path. However, data as taught by Baum before being placed on the T1 line is in analog from and transmitted on the telephone lines 40, 42, and 44. The calls from the call originators are then converted to digital signals (calls are digitized) and placed into the T1 line (see col. 8 lines 1-7). Thus Baum suggests that such calls are converted between digital and analog.

Applicant also argues that Baum is not a teaching of the communication of voice and that the mere teaching of Baum for “conversion between digital and analog” cannot

be stretched so far as to teach conversion of analog voice signals to/from digitized voice information. The Examiner respectfully disagrees since Baum suggests call signaling is for voice service (col. 19 lines 30-31) and the signaling bits (for T1 DS0) translate directly to E and M signaling (signaling for voice service) (col. 19 lines 65-66). Thus, Baum provides signaling for voice communication.

13. On pages 22-23 of the Applicant's remarks regarding claims 39 and 40, Applicant argues that Baum does not disclose parameters for configuring the information transmission device comprises information related to telephone signals generated by the information transmission device. The Examiner respectfully disagrees because, as cited in the independent claim 22, control signals are exchanged between the network application module and the modem for configuring the modem (col. 5 lines 50-60). The control signals are translated into multifrequency tones (col. 24 lines 52-53); in another word, the tones are produced by translating the control signals. The modem then decodes or converts the tones into DNIS in which the DNIS is correlated to the particular parameter (col. 25 lines 3-5, 19-20). Thus the parameter information is related to the multifrequency tones. Therefore Baum discloses this aspect of the claimed invention.

14. On pages 23-24 of the Applicant's remarks regarding claim 45 as previously discussed with respect to claim 22, please see response to claim 22 above.

15. With regard to independent claims 47 and 60, these claims recite similar features to those of claim 22, thus the rejections for these claims have been maintained for similar reasons.

16. Regarding other dependent claims, the rejections for these claims have also been maintained based on the reasons set forth above.

***Conclusion***

17. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to HOANG-CHUONG Q. VU whose telephone number is (571) 270-3945. The examiner can normally be reached on Monday through Thursday 8:30 AM to 6:00 PM EST. and alternate Fridays.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, AYAZ R. SHEIKH can be reached on (571) 272-3795. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/H. V./ 12/28/2009
Examiner, Art Unit 2476
/Ayaz R. Sheikh/
Supervisory Patent Examiner, Art Unit 2476